

What is claimed is:

1. A frequency domain interpolative CODEC system for low bit rate coding of speech, comprising:

a linear prediction (LP) front end adapted to process an input signal providing LP parameters which are quantized and encoded over predetermined intervals and used to compute a LP residual signal;

an open loop pitch estimator adapted to process said LP residual signal, a pitch quantizer, and a pitch interpolator and provide a pitch contour within the predetermined intervals; and

a signal processor responsive to said LP residual signal and the pitch contour and adapted to perform the following:

provide a voicing measure, said voicing measure characterizing a degree of voicing of said input speech signal and is derived from several input parameters that are correlated to degrees of periodicity of the signal over the predetermined intervals; extract a prototype waveform (PW) from the LP residual and the open loop pitch contour for a number of equal sub-intervals within the predetermined intervals;

normalize the PW by a gain value of said PW;

encode a magnitude of said PW; and

2. A system as recited in claim 1, wherein said predetermined interval comprises a frame.

3. A system as recited in claim 2, wherein said frame is preferably 20 ms.

4. A system as recited in claim 1, wherein said extraction of said PW sub-frame is preferably performed every 2.5 ms.

5. A system as recited in claim 1, wherein a ratio is computed comparing the ratio of the energy of the nonstationarity component of the PW to that of the stationary component of the PW which is averaged over five PW subbands.

6. A system as recited in claim 5, wherein a nonstationarity PW subband measure is encoded using a six bit spectrally weighted vector quantization scheme.
7. A system as recited in claim 6, further comprising:
reconstruction of a PW phase at a decoder for every said subinterval by separately generating said stationary and nonstationary PW components using the following:
 - a received PW magnitude;
 - a voicing measure;
 - said PW subband nonstationarity measure; and
 - said pitch frequency contour information.
8. A system as recited in claim 7, wherein said stationary component of said PW phase is reconstructed at a decoder for every said subinterval using a weighted combination comprising the following:
 - a previous PW phase vector;
 - a random phase perturbation; and
 - a fixed phase vector obtained from a voiced pitch pulse.
9. A system as recited in claim 8, wherein relative weights for said stationary and nonstationary components are determined by
 - a received voicing measure; and
 - said PW subband nonstationarity measure.
10. A system as recited in claim 9, wherein a rate of randomization of a random phase perturbation of said PW is controlled by a pitch frequency contour.

11. A system as recited in claim 10, wherein a range of said random phase perturbation is controlled by said received voicing measure and said PW subband nonstationarity measure.
12. A system as recited in claim 11, wherein said reconstructed stationary component of said PW magnitude and PW phase model is further processed every subinterval.
13. A system as recited in claim 12, wherein said further processing further comprises:
- low pass filtering said reconstructed stationary component to reduce excessive variations and to extract a stationary component of the PW; and
- preserving the PW magnitude after said filtering process.
14. A frequency domain interpolative CODEC system for low bit rate coding of speech, comprising:
- a linear prediction (LP) front end adapted to process an input signal providing LP parameters which are quantized and encoded over predetermined intervals and used to compute a LP residual signal;
- an open loop pitch estimator adapted to process said LP residual signal, a pitch quantizer, and a pitch interpolator and provide a pitch contour within the predetermined intervals;
- a signal processor responsive to said LP residual signal and the pitch contour and adapted to perform the following:
- provide a voicing measure, said voicing measure characterizing a degree of voicing of said input speech signal and is derived from several input parameters that are correlated to degrees of periodicity of the signal over the predetermined intervals;
- extract a prototype waveform (PW) from the LP residual and the open loop pitch contour for a number of equal sub-intervals within the predetermined intervals;

normalize the PW by a gain value of said PW;
encode a magnitude of said PW; and
reconstruct a nonstationarity component of a PW phase at a decoder every subinterval using only a received PW magnitude, a stationary component of said PW, said voicing measure, a PW subband nonstationarity measure and a pitch frequency contour information.

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15. A system as recited in claim 14, wherein reconstruction of the nonstationary component of said PW phase further comprises:

construction of a weighted mixture of the reconstructed stationary component of the PW phase and a noise component having the same energy as said reconstructed stationary component.

16. A system as recited in claim 15, wherein said weights are determined by said received measure and a frequency of a harmonic.

17. A system as recited in claim 16, wherein to achieve a range of frequency responses to realize a range of degrees of nonstationarity adjustment of poles of a high pass filter comprises a function of said received voicing measure and said frequency of the harmonic.

18. A system as recited in claim 17, wherein said high pass filtering of said weighted measure ensures higher rates of evolution and extraction of said nonstationary component of said PW.

19. A system as recited in claim 18 further comprising:

construction of a complex PW using a weighted sum of said reconstructed stationary and nonstationary components.

20. A system as recited in claim 19, further comprising:

restoration of relative levels of said nonstationary and stationary components as measured over five subbands.

21. A system as recited in claim 20, wherein said relative levels are transmitted by an encoder to said decoder as a nonstationarity measure.

22. A system as recited in claim 17, wherein said PW magnitude is preserved after said high pass filtering.